LecNet2™

Sound System Design Guide

Innovative Hardware/Software for Automatic Sound Systems

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LecNet2 Overview

The LecNet2 product group introduces a powerful series of audio components and unique solutions for the design and installation of audio systems. This Design Guide provides basic information on LecNet2 components, automatic mixing, the mix-minus approach to loudspeaker zoning, and specific information on audio teleconferencing. A variety of sound system block diagrams are included as examples of different applications for LecNet2™ components.

Applications for LecNet2 products include worship centers, courtrooms, paging systems, training and conference centers, council chambers and hotels. A single LecNet2™ sound system can provide teleconferencing, sound reinforcement and multiple mixes for recording all at the same time. LecNet2 components will operate as stand-alone devices or as part of a larger integrated system in order to provide a multitude of functions in simple-to-use, cost-effective rack-mounted assemblies.

USB and RS-232 compatible interfaces allow connection with all LecNet2 components, providing the supplied control software to communicate with each component. Individual function settings and signal routing can be customized for a particular application during setup, recalled from various screens during operation, or recalled by other brands of remote control systems. The LecNet2 serial port is completely compatible with control systems from AMX®, Crestron®, and any other equipment with RS232 interface compatibility.

In addition, a DANI (Digital Audio Network Interface) is provided so that the digital audio outputs of the master and slave units can be connected in stacked configuratons for larger applications.

LecNet2 Remote Control

There is much more to the story than the hardware. From the start we wanted to outfit these new products with a truly simple, robust and flexible remote control capability. The goal was to make life easier for third party control developers and let them focus on designing great control applications, rather than wrestling with the underlying protocols. The result is the LecNet2 Control System consisting of a streamlined command protocol that is human readable and transport neutral.

- Two high speed communication ports in every Lecnet2 device, USB and RS-232
- Visual command monitor built into many LecNet2 devices
- PC hosted Command Terminal program for easy testing and debugging
- PC hosted Net Server program which allows control of LecNet2 devices over a network connection

The command protocol is designed to be compatible with existing remote control platforms yet easy to program for. LecNet2 includes helpful tools like the command terminal program which allows a human to communicate with a LecNet2 device interactively. It is used to send commands and view the responses, much like working at the command prompt of a computer. Since the LecNet2 protocol has a clean, human readable syntax the commands and responses are easy to understand and debug. Incoming commands can also be viewed on the LCD screen of LecNet2 devices for troubleshooting purposes. LecNet2 is a system with high visibility into the command protocol.

The nature of the LecNet2 protocol makes it “transport neutral”, allowing it to be carried over a USB connection, an RS-232 link, or a network connection using the HyperText Transport Protocol (HTTP). This offers great flexibility in designing remote control applications. LecNet2 devices ship with both an RS-232 port and a USB port, with the operation of the Lecnet2 command protocol identical over each. Network connections to LecNet2 devices are possible using the PC hosted LecNet2 Net Server program. In either case complete control is possible by sending commands using the HTTP protocol, with even a humble web (HTML) page capable of simple functionality. LecNet2 is an open system for which controllers are easy to implement.

It is worth noting that the command terminal program is capable of accessing LecNet2 devices using any of the three connection types mentioned above, and may be used as a remote control and configuration tool for all LecNet2 devices. We think you will agree - LecNet2 offers unprecedented flexibility and convenience for remote control developers and system designers.

Automatic Microphone Mixing Algorithm

Sound reinforcement systems with multiple microphones and a distributed loudspeaker system are the rule in conference rooms, training rooms and boardrooms. In many cases the room has a low ceiling, which increases the acoustic coupling between loudspeakers and microphones, creating a significant challenge to providing adequate gain for sound reinforcement without acoustic feedback. In larger rooms with longer reverberation times, reinforced sound is re-circulated through multiple open microphones which can severely reduce the intelligibility of the overall sound system.

Automatic microphone mixers are an effective tool to minimize the effects of multiple microphones. All automatic mixers seek to open only those microphones which are being spoken into at any given time. Keeping the number of open microphones to a minimum reduces recirculated sound to improve intelligibility and eliminate acoustic feedback.

Automatic microphone mixers attenuate unused microphones following the rule that the gain applied to all open microphones is distributed among them so that it is always equal to a single open microphone. This process is commonly referrred to as NOM = 1, or the **N**umber of **O**pen **M**icrophones = 1. Following this rule, a sound system will perform the same with multiple microphones as it does with a single microphone with respect to feedback stability and intelligibility.

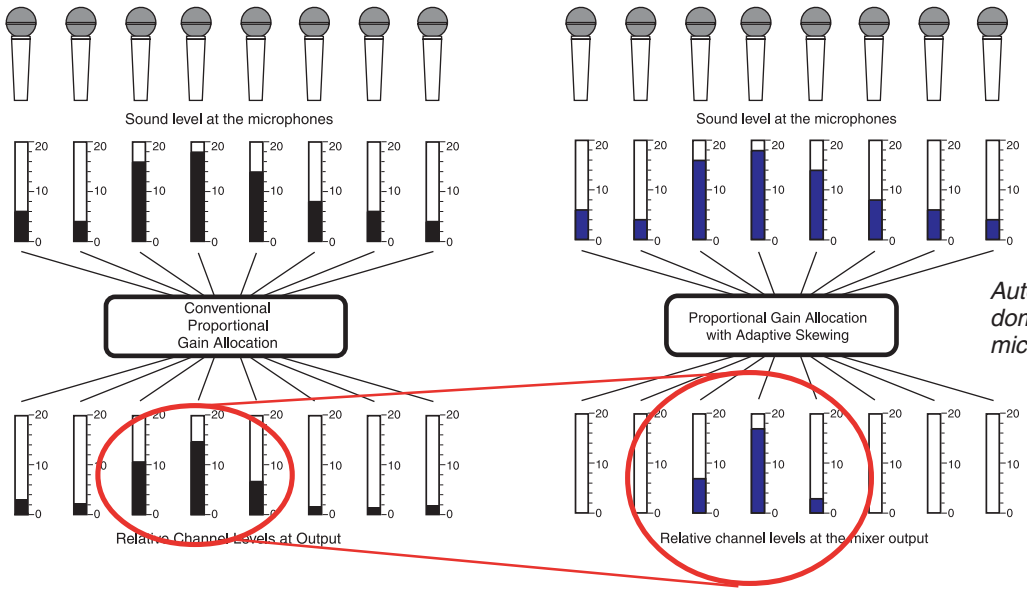
While all automatic mixers turn microphones on and off and implement some form of NOM attenuation, they are not all equally effective. The Lectrosonics automatic mixers employ a patented *proportional gain algorithm** to distribute gain across all channels in a seamless manner. An overall reference level is created by summing all channels. Then, each individual channel is compared with the overall reference level, and attenuated by the difference between its level and the reference level. The channels with the highest microphone signal levels thus receive proportionally higher gain than inactive or less active microphones.

NOM = 1 attenuation along with an adaptive threshold is inherent in this algorithm. As a result, accurate setup is as easy as using a standard mixer. And by using this continuous gain modulation technique, abrupt level changes and other anomalies normally generated by a switching or gating method are eliminated.

An additional feature of the Lectrosonics automatic mixing algorithm is an “intelligent” method of keeping track of which channel has been the loudest for the longest time period and skewing a “priority” toward that channel in the mix. This **AutoSkew™** process gives the priority channel up to 6dB of additional gain over the other channels making it appear to be more dominant in the gain-sharing allocation. The skewing rate is damped to avoid abrupt level changes that might be audible.

The AutoSkew™ algorithm also keeps non-speech transient sounds (coughs, bumps, clicks and pops) from affecting the gain allocated by the auto mixing process. For example, if someone bumps an unused microphone while someone else is talking, the gain of the microphone in use will not change.

AutoSkew™ is especially important in sound systems where the talker may be in close proximity to more than one microphone. For example, in a boardroom where multiple microphones are placed next to one another along a table, it is very common that a talker leans one direction or the other and is momentarily equidistant between two microphones. If both microphones were open and mixed at the same level, very audible comb filtering would occur. Another example would be a worship center where a person using a wireless lapel microphone approaches a gooseneck podium microphone and the voice is picked up equally in both microphones. AutoSkew™ reduces or eliminates comb filtering by not allowing any two or more channels to be mixed at the same level.



AutoSkew™ increases the dominance of the most active microphone channel.

*US Patent 5,414,776

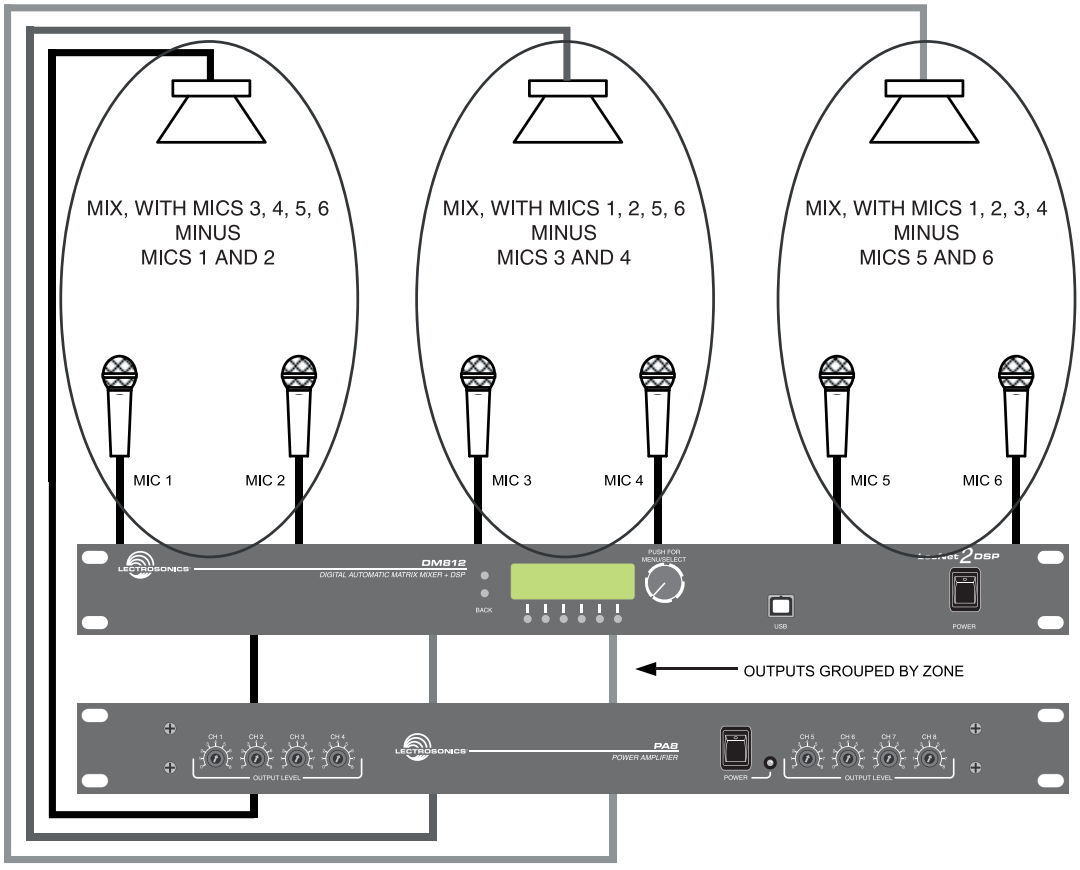
Mix-Minus Loudspeaker Zoning

Sound reinforcement systems installed in rooms with low ceilings often use multiple ceiling-mounted speakers distributed throughout the room in order to provide even coverage. When multiple microphones are used, as in a conference room, achieving any significant sound system gain before feedback can be difficult. Since system microphones will almost always be in the direct sound field of one or more of the distributed loudspeakers, feedback is virtually assured. Automatic mixers help the situation by minimizing the number of open microphones. Even with an automatic mixer, however, there may still be a need for some form of loudspeaker control in order to get acceptable GBF.

The Mix-Minus approach to loudspeaker zoning eliminates both of the problems mentioned above. Individual outputs from each microphone channel are delivered only to loudspeakers located far enough away from the microphone to eliminate feedback. In essence, the microphones and loudspeakers are physically decoupled. Mix-Minus systems using automatic mixers are even more stable against feedback, since the automatic NOM attenuation in the mixer reduces the gain on unused microphones. Mix-Minus routing does not change the way an automatic mixer operates, and in combined units such as the **DM Series** matrix mixers, all of these functions are seamlessly integrated.

As an example, consider a conference room with multiple microphones and ceiling speakers. Assume the average distance from a microphone to its closest loudspeaker is 6 feet. Using a DM matrix mixer, a Mix-Minus feed can be generated for each loudspeaker that does not include the microphones close to each specific loudspeaker. If, for example, the next closest microphone to a loudspeaker is on the order of 12 feet away, the sound system will have picked up 6dB more gain before feedback. In difficult acoustic circumstances, 6dB may be the difference between a functioning sound reinforcement system and an expensive problem.

DM Series matrix mixers combine the elegance of the LecNet2 automatic mixing algorithm combined with the feedback reduction and stability of Mix-Minus. This design architecture provides an outstanding foundation for sound systems to simultaneously provide reinforcement, teleconferencing and recording. Mix-Minus also reduces echoes heard at the far end of a teleconference, while full duplex operation is preserved



Digital Matrix and DANI™ Bus

The core of the DM Series processors consists of a digital matrix and a digital bus called DANI (digital audio network interface). The digital matrix is common to all units in a system. The DANI bus interconnects the hardware to allow access to the matrix signal flow and transfer data required for automatic mixing functions. In order to understand the power and functions available with this architecture it is helpful to think of them as entities separate from the hardware.

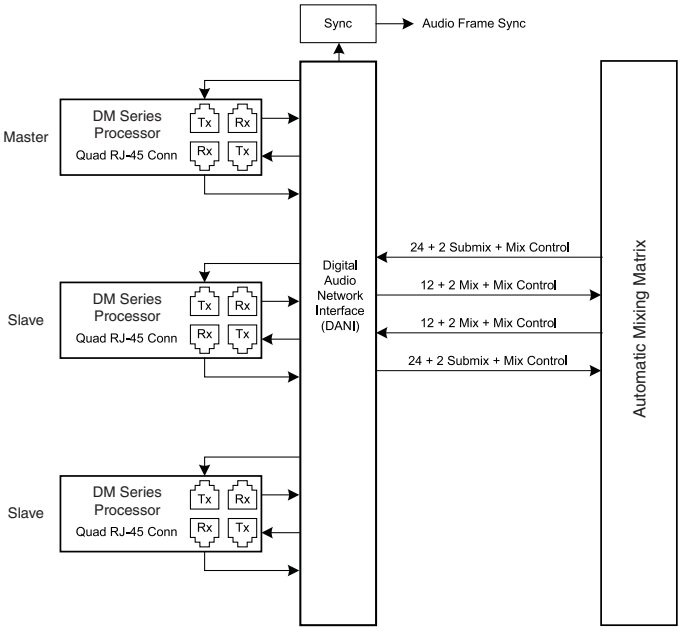
In this sense a DM processor is simply a hardware-based tap into the digital matrix via the DANI bus to interface various types of microphones and audio equipment with the digital matrix. Thus connected, the processors distribute audio signals and share information about each input and output to provide a myriad of features and functions.

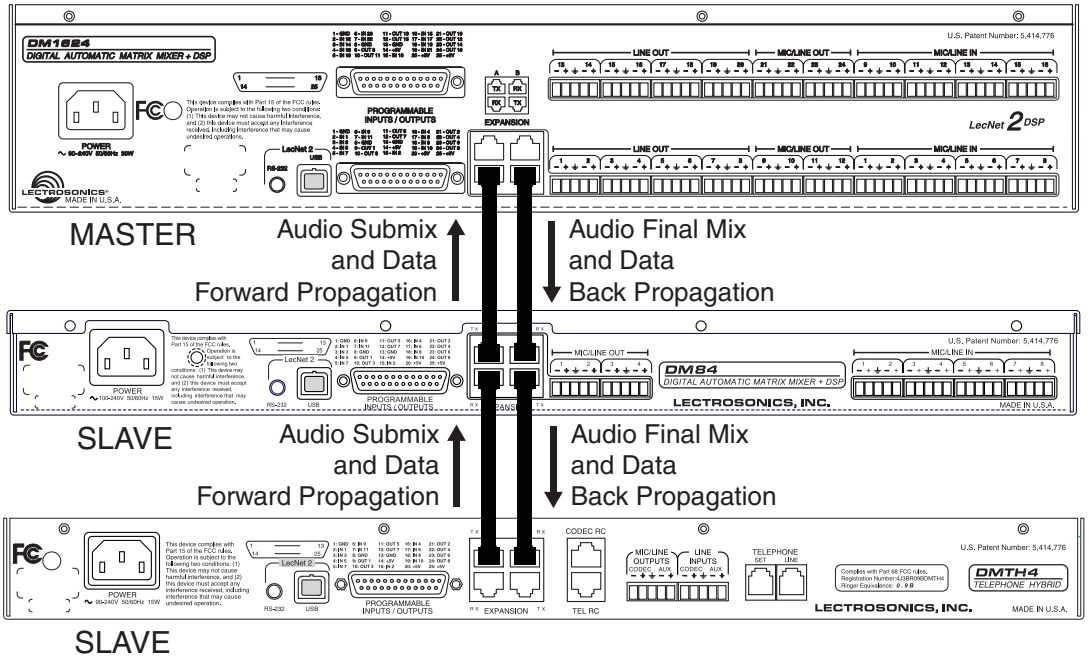
When multiple DM processors are stacked, each unit participates with the digital structure in several ways:

- Delivering audio signals from its input terminals into the forward-propagated submix bus
- Passing back-propagated final mix signals from the unit above it to the next unit below it
- Applying gain and signal processing to the audio signals at its input terminals
- Delivering audio signals to its output terminals as selected by the setup
- Applying signal processing to the signals routed to its output terminals
- Receiving and transmitting data required for the automatic mixing process in the matrix

The digital matrix is common to all processors in the stack, with automatic mixing taking place at the crosspoints in the digital matrix. The output of each crosspoint is then available at a variety of output terminals on various processors in the stack.

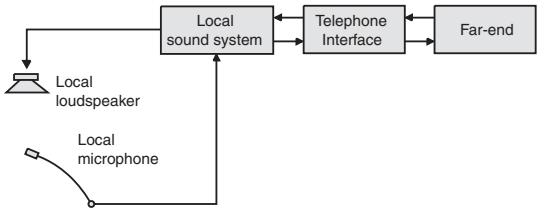
Different processor models interface with the digital matrix in different manners. Audio signals and data are propagated from the Slaves to the Master unit in a stack, then the data and some of the final mix signals in the Master are back propagated to the Slaves. This provides additional final mix outputs at the output terminals on the Slave units.





Teleconferencing with DM Series Processors

The fundamental problem in teleconferencing with a sound system is microphone/speaker acoustical coupling as is illustrated below. Far end audio is delivered by the loudspeakers in the room and the microphones pick it up and return it to the far end. The delay through this process creates an echo heard on the far end.



There are several methods used to reduce or eliminate the echo heard on the far end of the conversation:

- Optimal design in the sound system to minimize the coupling between loudspeakers and microphones.
- Mix-minus matrix routing.
- Automatic microphone mixing.
- Digital acoustic echo cancelling.

Matters become more complex when the sound system is required to provide both teleconferencing and sound reinforcement. A gain proportional automatic mixing process is widely recognized as the optimum solution for sound reinforcement, but it places significant demands on an acoustic echo canceller used for teleconferencing.

The matrix mixer enables complex signal routing and level controls without limitations. The matrix mixing allows **"mix-minus"** zoning of microphones and loudspeakers to decouple them and reduce or eliminate acoustic feedback and echoes. NOM attenuation is applied by the DSP at the crosspoints in the matrix, which essentially provides 24 separate automatic mixers, each with its own NOM mixing bus. Four different mixing modes can be selected at the crosspoint for each input, so each input can participate differently in each output mix.

The **automatic mixing** process uses a seamless algorithm that eliminates gating and its ill-effects. Gain is proportioned among all inputs assigned to each output channel in a seamless and continuous manner based upon microphone activity. The algorithm incorporates an adaptive AutoSkew™ process to eliminate artifacts such as comb filtering and abrupt gating that occur with conventional automatic mixing schemes. Audio from the far-end of a conference participates in the local mixing algorithm just like a microphone in the local sound system.

Two digital **acoustic echo cancellers** are provided in the DMTH4 to further reduce the return of local signals to the far-end. One operates on the telco connection and the other is dedicated to the video codec connection. In conjunction with the automixing process, echoes are minimized and not heard at the far end.

ERL

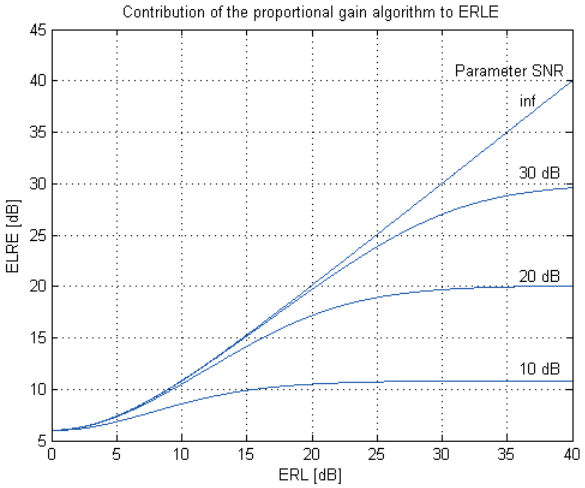
ERL (echo return loss) refers to the natural attenuation of the far-end audio signal as it circulates from the far-end through loudspeakers and microphones in the local sound system and back to the far-end. Good design in the local sound system will reduce the acoustic coupling between loudspeakers and microphones using physical placement and mix-minus matrix routing. Depending upon room size and acoustics, it is often impossible to achieve adequate decoupling to avoid an echo heard by the far-end during a teleconference. Thus, other types of processing are needed to further reduce the return echo.

ERLE

ERLE (echo return loss enhancement) refers to additional circuits and processes used to further increase ERL. Common methods are to use automatic mixing and digital echo cancellation.

Return Loss Enhancement

The gain proportional automatic mixing algorithm* in the DM Series processors not only provides seamless mixing for local sound reinforcement without abrupt gating, but it also contributes significantly to ERLE. The additional contribution is plotted in the following graph.



Digital echo cancellation is another method of reducing the echo delivered to the far-end. The concept, described in very simple terms, is to have the DSP recognize the far-end audio and subtract it from the transmitted audio to remove any echo they might hear at the far-end. Sounds simple, but in a sound system with multiple microphones and loudspeakers, it is not easy to identify the far-end audio in the complex mix of local sound, local noise and the effects of the room on the far-end audio delivered by the local loudspeaker system. When there is no sound or noise in the local room, the DSP can do a decent job of identifying the far-end audio and subtracting it from the transmitted signal, but this is rarely the case in full duplex teleconferencing.

In a simple sound system arrangement, the local microphone can be muted when nobody is talking in the local room. A simple gated mixer can provide this function. With no open microphones locally, there is obviously no return echo signal. This requires that a threshold level be set high enough to keep the microphone from being opened by background noise, but low enough to allow it to open when someone speaks. When the local microphone is open, a return echo path is created, which is when a DSP echo canceller is needed. Given the wide variety of human voices and the dynamics of noise in a meeting room, a gated mixer is often not the best choice.

Using a dedicated DSP echo canceller on each input of the local mixer (referred to as "distributed echo cancellation") is an expensive but effective approach to reducing the return echo. The process requires the algorithm to "converge," which is to identify the far-end audio and subtract it from the signal sent to the far-end. This requires at least a brief moment when there is very little local sound or noise, with significant far-end audio present in the room. If nobody moves and there are no gain changes made to local microphones and loudspeakers, it is possible (in theory) to effectively remove return echo, but this is not a very realistic situation.

The theory behind distributed echo cancelling is that once the DSP has converged, it can continue to subtract far-end audio even when the local microphone is open and far-end audio is present at the same time. If there are any changes in gain, noise or acoustics in the local space and equipment, the DSP must re-converge, which requires another brief moment with little or no local noise or sound, and significant far-end audio present.

A gated automatic mixer does not change the gain when the microphone is open, it just turns the channel off and on abruptly. This helps with distributed echo cancelling since the microphone is completely muted when not in use, but it is very "choppy" sounding in the local sound reinforcement system.

A gain proportional automatic mixer applies the most gain to the most active microphone with smooth, continuous changes. This makes it extremely effective for local sound reinforcement, but the continuous gain changes make it difficult for the echo canceller to remain converged and effectively reduce the echoes at the far end.

The DMTH4 in conjuction with a DM Series processor offers a unique approach to the problems with simultaneous teleconferencing and sound reinforcement. The patented adaptive gain proportional mixing algorithm works in conjunction with a centralized echo canceller to address a variety of issues. The automatic mixer provides seamless allocation of gain to local microphones through a mix-minus matrix to reduce background noise and decouple loudspeaker and microphones, while a very fast converging DSP echo canceller operates on the composite transmitted signal being sent to the far end. This combination of processes is possible only with the latest DSP technology.

The auto mixing algorithm adapts to changes in background noise continuously, and unlike a gated mixer there are no threshold levels to adjust. A sum of all channels is the reference signal, each channel level is compared to this reference and the individual channel gain is adjusted to apply NOM attenuation. Gain is adjusted continuously to eliminate audible artifacts that gating and abrupt level changes can cause. As the common mode noise in the room changes, all channels are affected equally. The end result is seamless, adaptive auto mixing that requires no calibration or threshold adjustments.

Each individual output of the matrix operates as a separate NOM bus, so a particular input can be assigned to multiple outputs with mix parameters adjusted differently for each output. In other words, gain and mix mode are configured independently for each matrix crosspoint, resulting in great flexibility. Four mix modes are supported: Auto, Direct, Override and Background.

The echo canceller converges continuously when the level of the far side signal exceeds a minimum level, and the ratio of the far side signal to local room sound exceeds a minimum ratio. This dynamic control prevents divergence during periods of silence from the far side room or in "doubletalk" situations. The convergence takes place very quickly to keep up with the changes made by the automatic mixing algorithm and other changes that occur in the room. Setup is greatly simplified and any adjustments, such as level changes made with a remote control system, are accommodated automatically.

The convergence speed is adjustable in the control panel GUI to fine tune it to a particular situation. Faster convergence times can track changes in the room almost instantaneously, but the depth of echo cancellation will be reduced. Slower convergence times take a bit longer to fully converge, but produce greater echo cancellation. The ERLE value achieved by the echo canceller is displayed on the GUI and the effects of altering the convergence rate will be immediately visible and audible.

An important final note on the DMTH4 is the fact that the echo canceller will never "diverge" (lose convergence). This unique algorithm will also converge on a continuous sine wave, which is especially important when DTMF tones are present in the room. Since the echo canceller will never diverge, there is no need for a "panic button" (as is used in other designs) to generate a noise burst to help the echo canceller re-converge.

Video Follow Audio - A Practical Primer

As teleconferencing advances and becomes more common, new design requirements have risen that can be addressed with the DM Series. In a large room, a single camera can only deliver an image of the entire table, leaving the far-side viewers to guess which of the little talking heads in the picture is actually speaking. Video follow audio methods allow the video signal to track the audio conversation. This is done via two possible control methods - pan/tilt cameras or multiple cameras through a video switcher. To do so requires a signal from the automatic mixer that lets the camera control system know to which zone to bring the camera.

The DM Series mixers have two ways of providing this data. If controlling the system through a third party device, the control system can acquire the information via the serial port. The programmer codes a looped inquiry that constantly polls the mixer for the status on the various microphones. When the microphone comes to within 6dB of full gain, the control system is notified via the serial port and either switches to the correct camera or calls up the new pan/tilt coordinates.

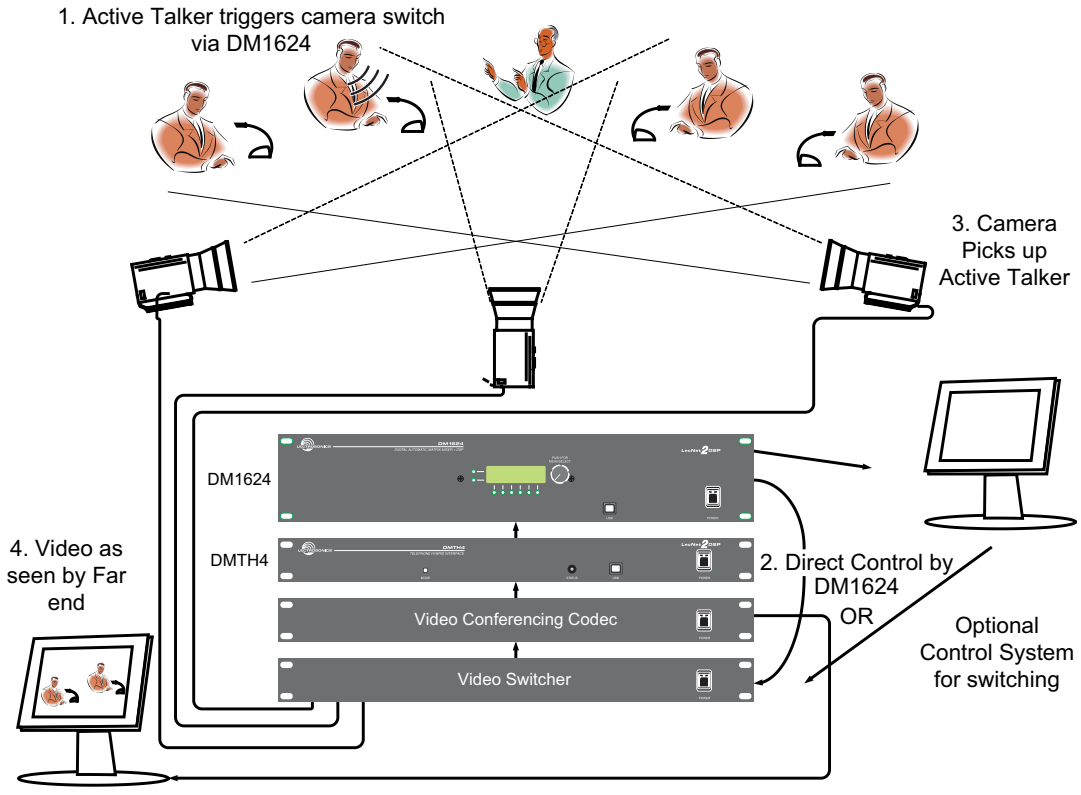
The second, and sometimes simpler, method is to set the programmable outputs of the DM Series to emulate a contact closure upon activity at any given set of microphones. Entire groups of microphones can be assigned to a single pin. The contact closure can then activate either the switcher or the pan/tilt platform.

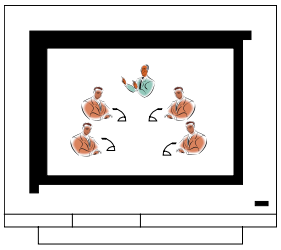
To smooth the switching activity, the DM mixers provide programmable parameters for both input **qualification time** and **hold time**. Qualification time is the time that the DM will withhold the notification of channel activity. This helps prevent a switcher from triggering when someone coughs, moves a paper, bumps a microphone or some other momentary sound occurs.

The Hold time keeps the contact closed if the microphone goes quiet, to avoid losing the camera simply because someone pauses briefly. Both parameters are adjustable from 0 to 25.5 seconds in 0.1 second increments to prevent false triggering or jumpiness in the video as it attempts to follow a conversation.

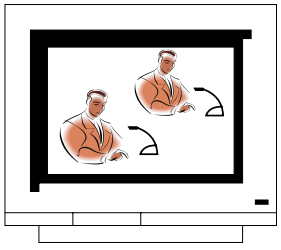
This leads to the other question before the designer: whether to use multiple cameras or pan/tilt platforms. That often depends on the nature of the conferences. For very large and/or very active rooms, where the conversation may be unstructured and free flowing, using multiple cameras will work better. Pan/tilt platforms, while reducing the number of cameras, fail to travel quickly enough during a lively discussion to offer easy tracking of the video to the audio. Given the lower costs of cameras compared to pan/tilt mechanisms, the multiple camera design makes better sense in many cases. If used in a smaller room or where the order of conversation is more structured (for example, Roberts Rules of Order in a council chamber), then a pan-tilt design can offer smooth transitions from one microphone location to the next, emulating the movement of one's head if they were observing the dialogue.

Please Note: Automated switching works best with systems of 5 or more microphones.


1. Active Talker triggers camera switch via DM1624
3. Camera Picks up Active Talker
DM1624
DMTH4
Video Conferencing Codec
Video Switcher
2. Direct Control by DM1624 OR
Optional Control System for switching
4. Video as seen by Far end



Single Camera View



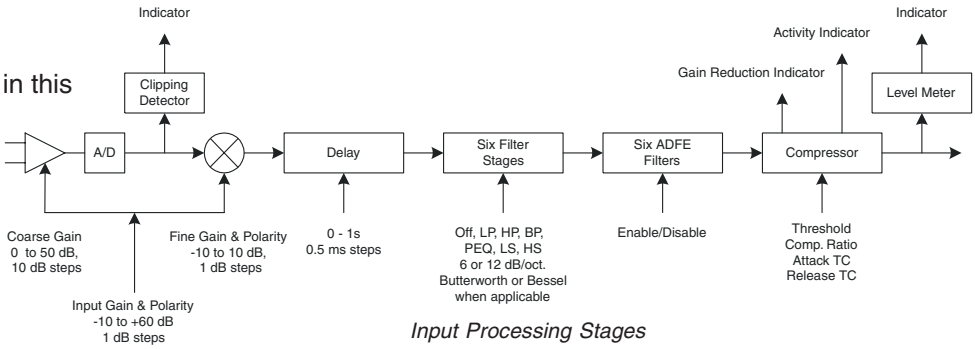
Video Follow Audio

DSP Features and Setup GUI

All models in the DM Series provide extensive digital signal processing to optimize each audio channel for its intended purpose in the system.

Input Signal Processing

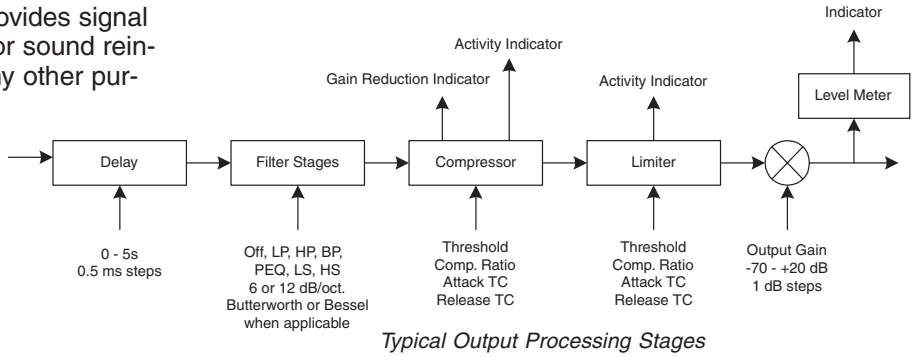
Signal flow for each input is shown in this diagram. Following the input preamp and level control are the processing stages for delay, six filter stages, six ADFE feedback eliminators and a compressor. Every function and feature on every stage can be fully implemented on every input channel since there is no limitation on the resources available.



Input Processing Stages

Output Signal Processing

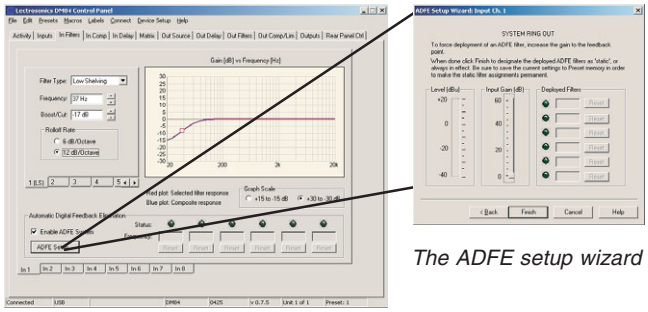
Every output on every DM processor provides signal processing block to idealize the signal for sound reinforcement, recording, media feeds or any other purpose. The signal processing chain is especially useful in the DMPA12 digital power amplifier. Back propagated final mix signals from the Master unit in the system are individually processed at the DMPA12 outputs, so the same signal mix from the Master can be used for recording, teleconferencing and sound reinforcement.



Typical Output Processing Stages (varies slightly by model)

ADFE (Automatic Digital Feedback Eliminator)

A special filter is included in the DSP stage at the inputs to eliminate acoustic feedback. With the processor setup and running, a setup wizard provides a simple procedure to identify and eliminate feedback. As the gain is gradually turned up, a slight oscillation begins to be audible (ringing). When the ringing begins, a very narrow notch filter is automatically deployed to eliminate the oscillation. The filter can then be stored in a Preset to make it permanent.

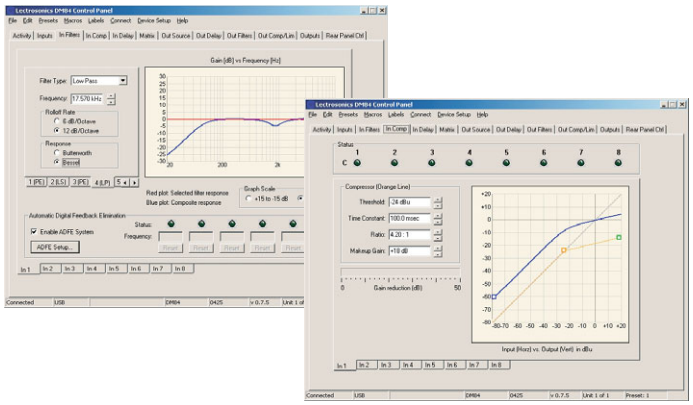


The ADFE setup wizard

Windows Tabbed GUI

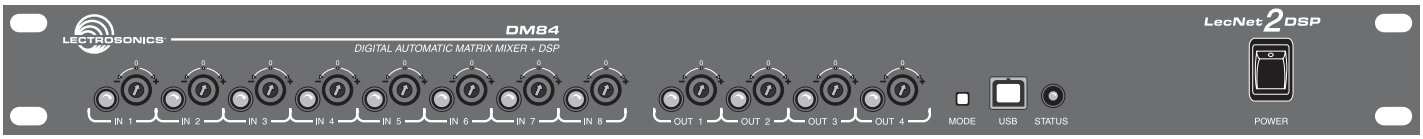
The graphical user inteface supplied with all LecNet2 products uses a familiar Windows® tabbed structure.* Tabs across the top of the screen open working pages to set up the parameters for each type of signal processing. Individual channels are selected and and other detail setup pages opened with a variety of buttons in the page display.

The GUI allows values to be entered directly, click and drag setup with the mouse, and click and hold buttons to scroll through available values. Setup is intuitive and visual displays illustrate the setup parameters as the data is entered.



** Windows® is a registered trademark of Microsoft, Corp.*

LecNet2 Audio Components

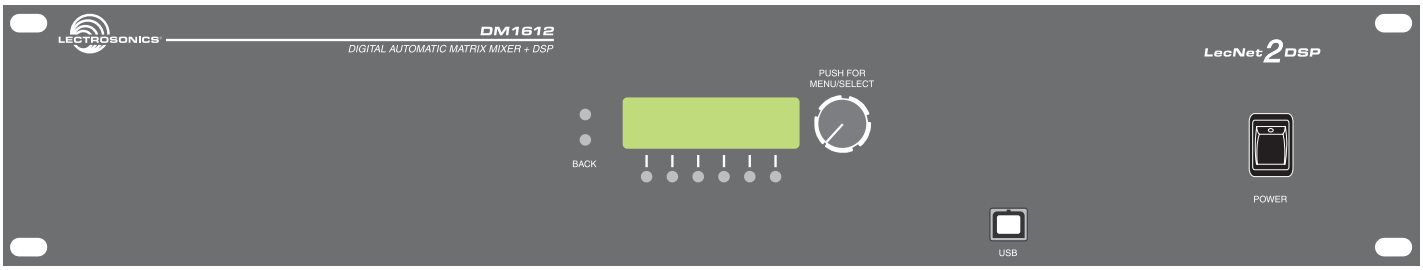


DM84

The DM84 is an 8-in/4-out digital matrix mixer allowing every input to be routed to any or all outputs. Automatic microphone mixing using a proportional gain algorithm allows for greatly increased intelligibility and gain before feedback. Each input can incorporate up to 6 filter stages plus compressor, ADFE and delay. Each of the 4 outputs provides a digital delay, up to 9 filters and a compressor/limiter. Front panel controls and indicators allow the DM84 to be used in a similar manner to the older AM8 and AM8/TC units.

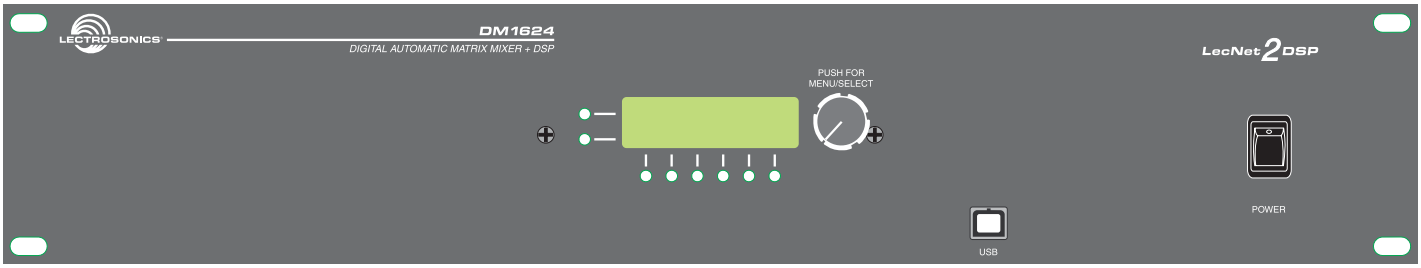
DM812

The DM812 is an 8-in/12-out digital matrix mixer allowing every input to be routed to any or all outputs. Automatic microphone mixing using a proportional gain algorithm allows for greatly increased intelligibility and gain before feedback. Each input can incorporate up to 6 filter stages plus compressor, ADFE and delay. Each of the 12 outputs provides a digital delay, up to 9 filters and a compressor/limiter.



DM1612

The DM1612 is a 16-in/12-out digital matrix mixer allowing every input to be routed to any or all outputs. Automatic microphone mixing using a proportional gain algorithm allows for greatly increased intelligibility and gain before feedback. Each input can incorporate up to 6 filter stages plus compressor, ADFE and delay. Each of the 12 outputs provides a digital delay, up to 9 filters and a compressor/limiter.



DM1624

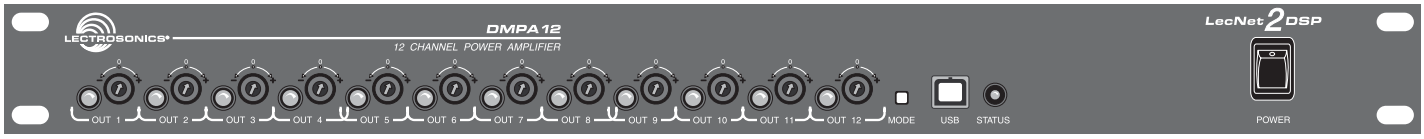
The DM1624 is a 16-in/24-out digital matrix mixer allowing every input to be routed to any or all outputs. Automatic microphone mixing using a proportional gain algorithm allows for greatly increased intelligibility and gain before feedback. Each input can incorporate up to 6 filter stages plus compressor, ADFE and delay. Each of the 24 outputs provides a digital delay, up to 9 filters and a compressor/limiter.

DMTH4

The DMTH4 integrates telephone lines, video codecs and external audio sources into the digital bus structure of DM Series processors so these sources operate as though they are another microphone or audio input in the sound system. The unit is much more than just a telephone interface. Instead, it is a complete DM Series digital matrix processor, with a 3-in/24-out digital matrix, automatic mixing and comprehensive signal processing on every input and output. Two acoustic echo cancellers are provided, one dedicated to the CODEC and the other to the TEL connection. An extremely fast echo cancellation algorithm converges so fast it tracks and adapts to the level changes in the auto mixing process, which allows centralized echo cancellation.

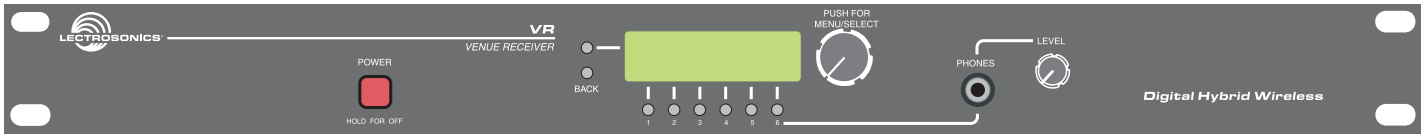
PA8

The PA8 analog power amplifier provides 8 discrete channels for use in multi-speaker sound systems. It is fully protected from open and shorted outputs and thermal overheating. Features include individual level controls on each channel, depluggable input and output connectors, passive cooling and a single rack space for installation in standard 19" racks. Adjacent channels can be bridged for added power output.



DMPA12

The DMPA12 is a digital 12-channel power amplifier and DSP processor in a single, 19" rack enclosure. Audio inputs are taken from the final mix signals back propagated via the DANI bus from the Master in the system. These final mix signals are then processed individually at each output channel in the power amplifier to apply digital delay, equalization filters, compression and limiting. The unit runs very cool with 10 Watts per channel using Class D amplifiers and passive cooling (no fan).



Venue Wireless Microphone Receiver

The Venue Receiver system is a modular UHF design that operates with Digital Hybrid Wireless™ transmitters, and a variety of analog transmitters. It consists of a Venue Receiver Master (VRM) and one to six plug-in receiver modules. The VRM includes an antenna multicoupler, computer communications interface and the mechanical rack mounting for the receiver modules. Supplied software allows the Venue receiver to be addressed via RS232 or USB as part of a LecNet2 audio system.

Wired Remote Controls

RCW-DMTH4

- Connects directly to the Remote Control Port (DB-9) on the DMTH4
- Brings the hybrid front panel controls to a table top surface
- LED indicators display status and volume adjustment activity
- Soft-touch switches
- Durable laser-engraved nomenclature
- Machined aluminum housing

The RCW-DMTH4 provides the main operating controls for the DMTH4 hybrid in a remote location. The circuit board is mounted in an attractive, powder-coat finished box. Soft touch switches and highly visible LEDs provide simple, intuitive operation and instant recognition of the status of the DMTH4. Two RCW units would be used if the DMTH4 is connected to both an analog line and a codec. This allows independent control of both sources during a bridges three-way (room/video/audio) conference.

RCW-TEL

- Remote Control for DMTH4 Digital Telephone Hybrid
- LEDs for Connect and Privacy status
- Push-button switches for remote volume control
- Fits single-gang conduit box
- Connects to a DB-9 connector on the DMTH4
- 7-conductor wiring with screw terminals

The RCW-TEL provides the main operating controls for the DMTH4 hybrid in a remote location. The circuit board is mounted to a single-gang wall plate. Soft-touch switches and highly visible LEDs provide simple, intuitive operation and instant recognition of the status of the DMTH4. Two RCW-TELs would be used if the DMTH4 is connected to both an analog line and a codec. This allows independent control of both sources during a bridged three-way (room/video/audio) conference.

RCW-VLS

Remote Level Control for Lectrosonics auto mixers, models: AM8, AM8TC, AM8/4, AM16/12, DM84, DM812, DM1612 and DM1624.

- LED indicates active status
- Screw terminal connections for reliability
- Fits single-gang conduit box
- Detented rotary action
- Selectable (by jumper) FULL or -15dB attenuation
- 3-conductor wiring

The RCW-VLS is a rotary volume control mounted on a circuit board and a single-gang wall plate for use with Lectrosonics automatic mixers and matrix units. The rotary action is detented for a smooth, accurate feel and repeatable selection. The unit can be connected to control individual channels in any combination or to the main output control point on units with VCA taps. It is also used as an analog control on units with Programmable Input connections. Fully clockwise, no attenuation is applied. Full counter clockwise, the level is either fully attenuated or is reduced to -15dB, as determined by an internal jumper on the circuit board.

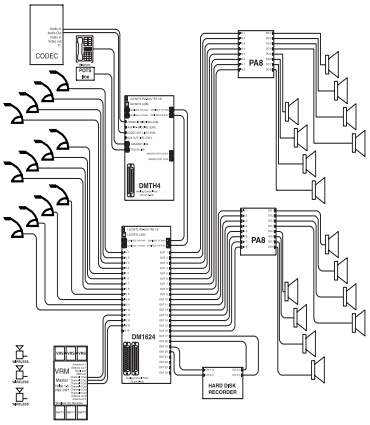
RCW-PB4

- Remote control for DM Series products
- 4 Momentary contacts
- 4 Indicator LEDs
- 8-Conductor wiring with screw terminals
- Fits single-gang conduit box
- Four customizable function labels

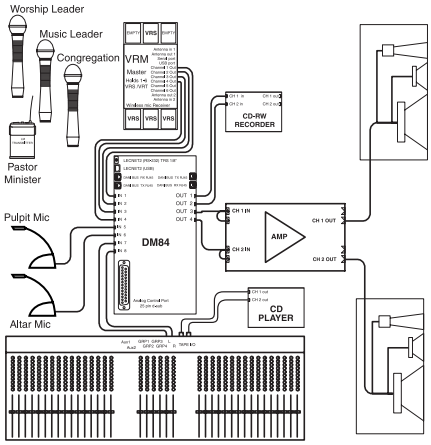
The RCW-PB4 is designed to provide remote control capability with any DM Series product by actuating macros, presets, or level controls. The RCW-PB4 connects via any small gauge multiconductor wiring to the DB25 connector on the DM series units. This control can be configured in the DM software to actuate virtually any function including (but not limited to) complete signal routing changes, group or individual level controls on inputs or outputs, or even change the nature of the automix functions. LED's can be programmed independently from the buttons for status indicators or user feedback.

Example Sound System Designs

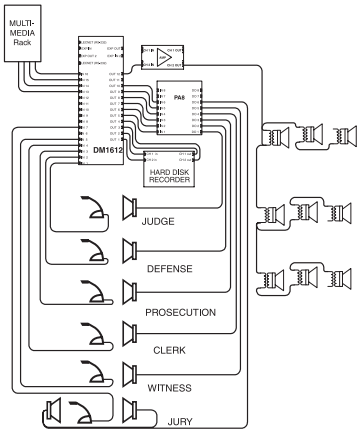
Boardrooms, Training and Conference Centers



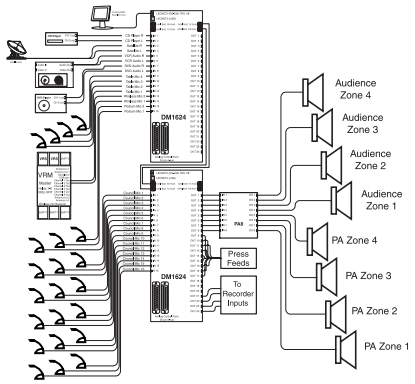
Worship Center Automated Sound Systems



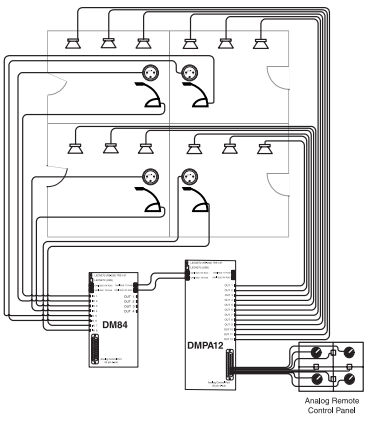
Courtroom Sound Systems



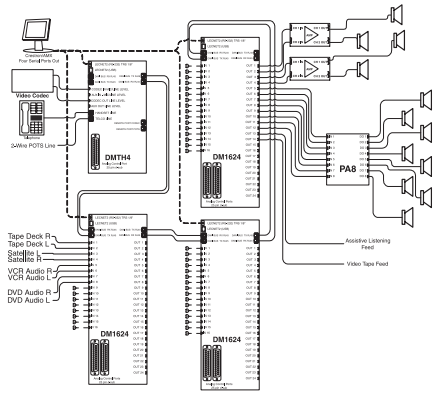
City Council Chamber



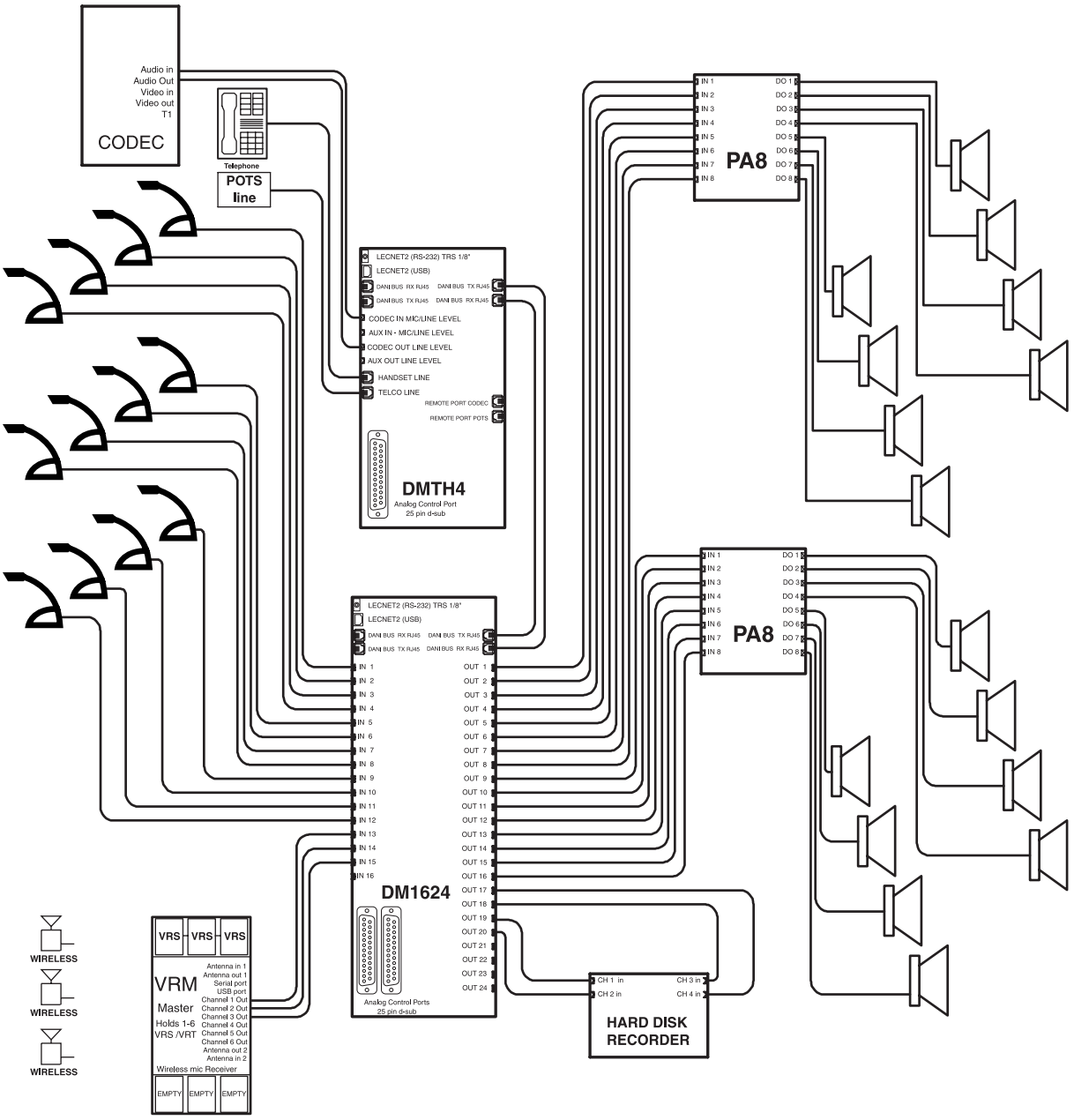
DM1612 in Multi-Room Combining



Collegiate Distance Learning/ Multi-Purpose Hall



Boardrooms, Training and Conference Centers

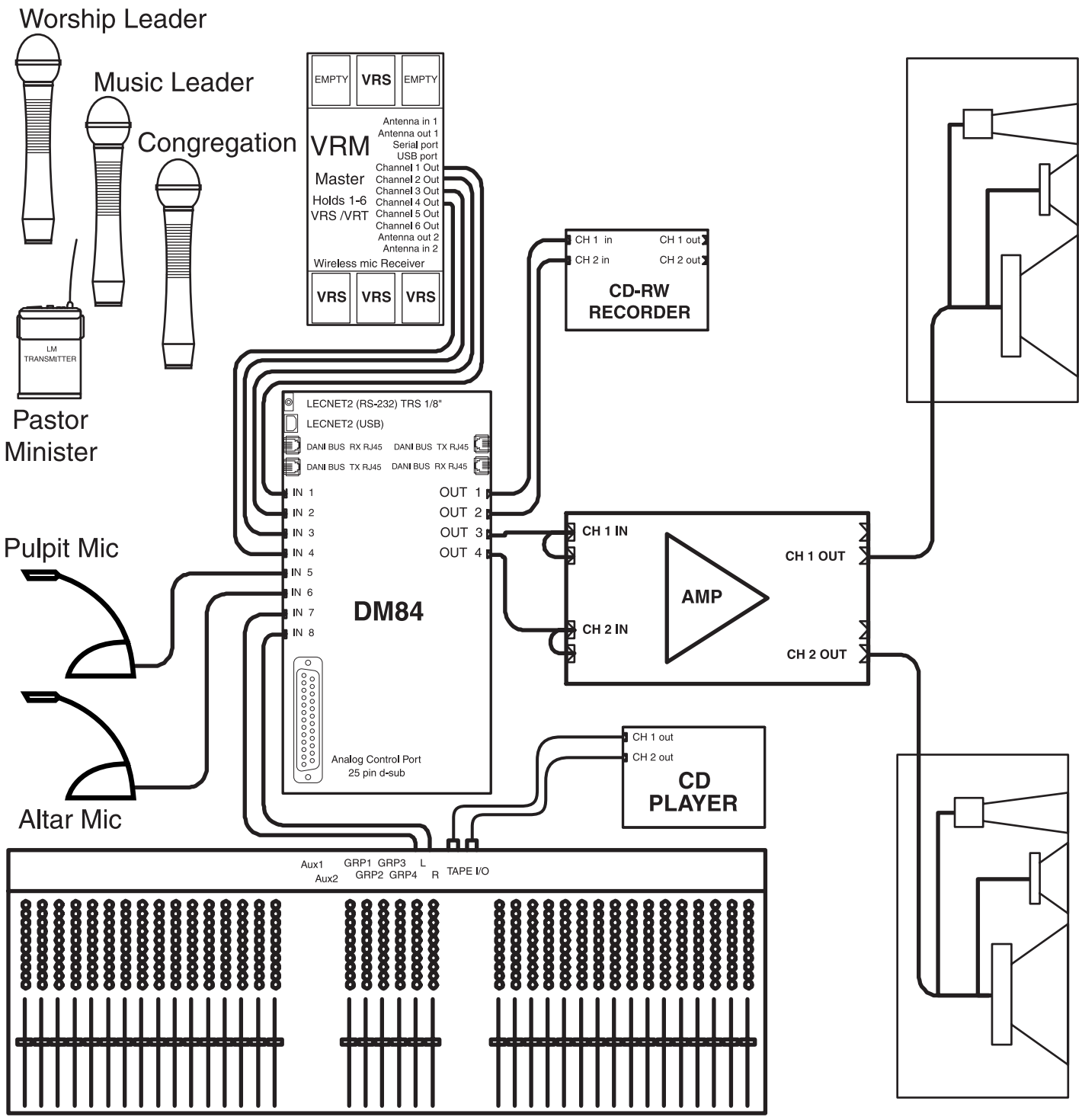


This diagram details a sound system appropriate for the boardroom or conference room. The installation includes both audio and video teleconferencing (which can be bridged for simultaneous 3-site conversations), automatic recordings of these conferences, tape input, control systems and speaker zoning. The DM1624 takes input and provides automixing from 12 wired microphones, three wireless bodypack systems, telephone lines, via the DMTH4, and a hard-disk media recording system. In teleconferencing modes, whether audio only or video, the audio from the teleconference is recorded automatically.

Due to the low ceilings typically founding in boardrooms, a mix-minus speaker routing system is recommended. Using the DM1624 matrix, 16 speaker zones have been set up. The DSP control within the DM1624 allows zone-specific equalization and feedback control. The 16 speaker zones are amplified with two PA8 amplifiers. Control of the system can be achieved with an AMX or Crestron control system. Programming can be easily altered on site by updating macros using the USB or RS232 ports and LecNet2 software.

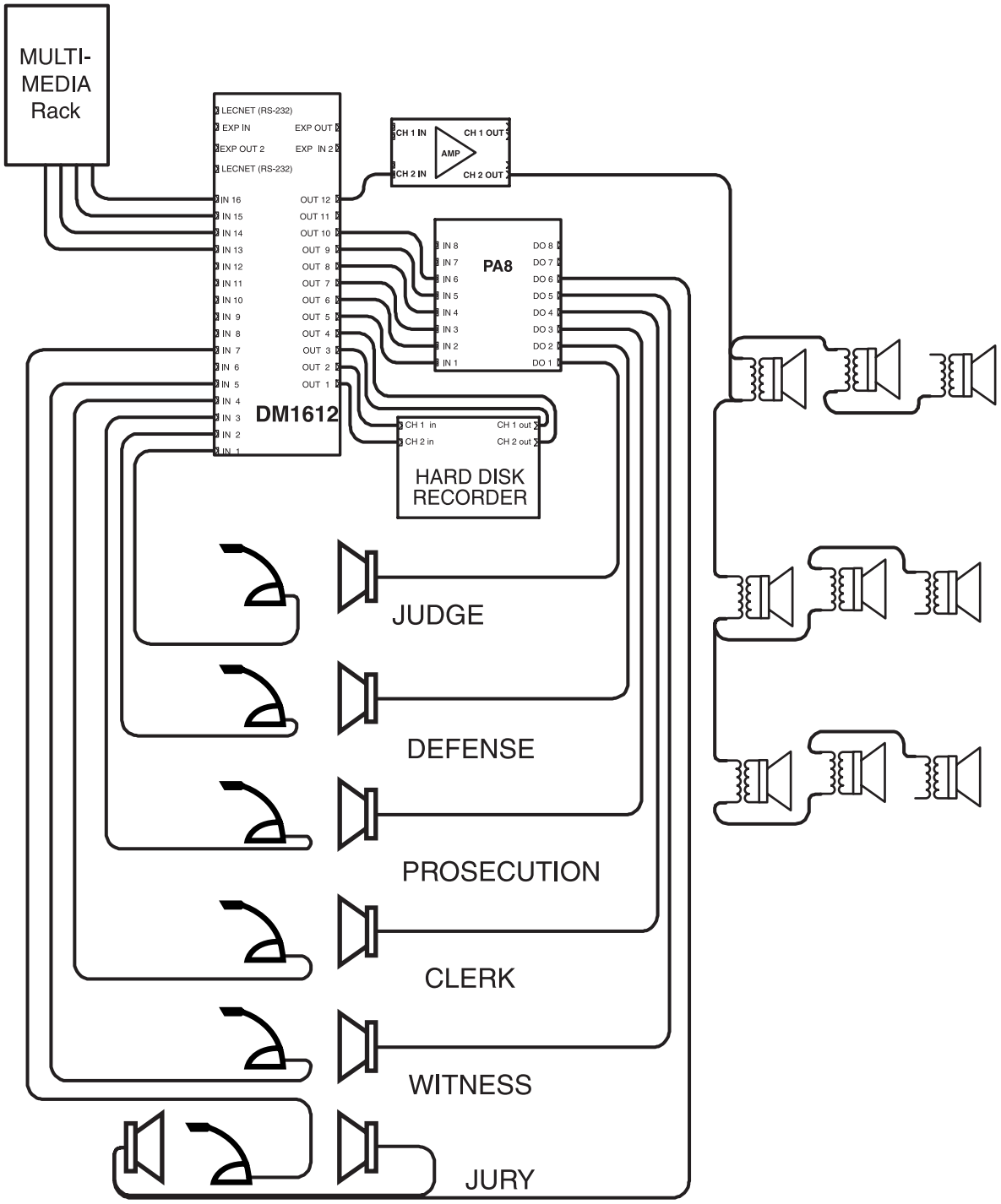
The wireless microphone system is a Lectrosonics Venue receiver loaded with three VRS modules. Bodypack transmitters can be LM, SM, UM400 or, by setting compatibility modes, any transmitters from the previous 200 series.

Worship Center Automated Sound Systems



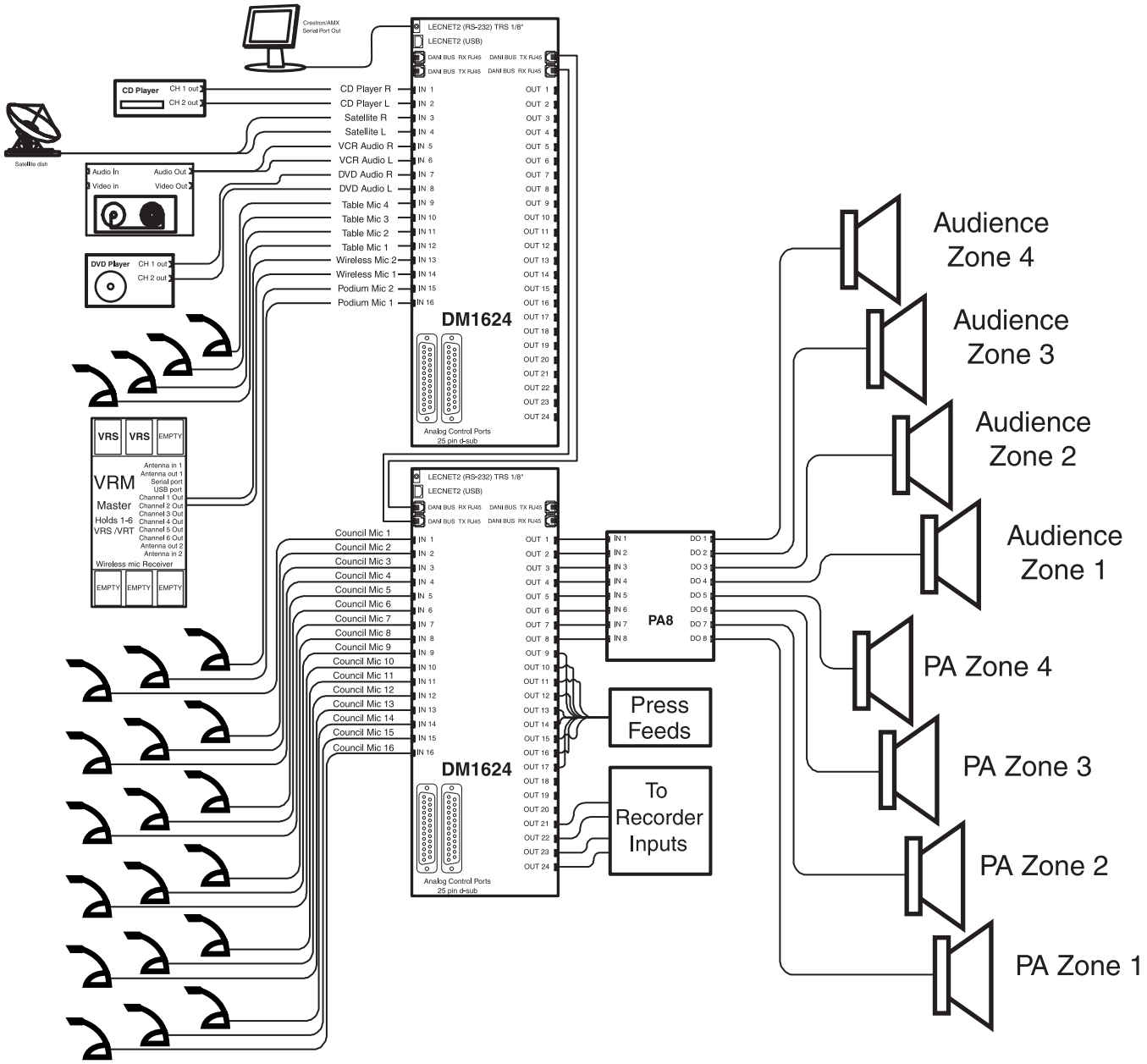
This diagram shows how the DM84 can be used as an automatic “front end” for the speech microphones in a church system. All of the speech microphones are passed through the DM84. The direct outs from the DM84 are then input to separate channels of the mixing console. Full control over the speech microphones is maintained by the console operator, but the DM84 eliminates the need for routine “microphone chasing.” The console operator is thus free to concentrate on other mixing and adjustment tasks. In addition, for worship services where no console operator is present (weddings, funerals, etc.), the addition of the DM84 to the console provides fully automatic operation.

Courtroom Sound Systems



This diagram shows a typical courtroom sound system, including microphones for each of the key people, an automatic mixing function, and a recording system. A 70 volt distributed speaker system is fed from a main output on the DM1612. A multi-media rack provides input signals from audio and video recordings.

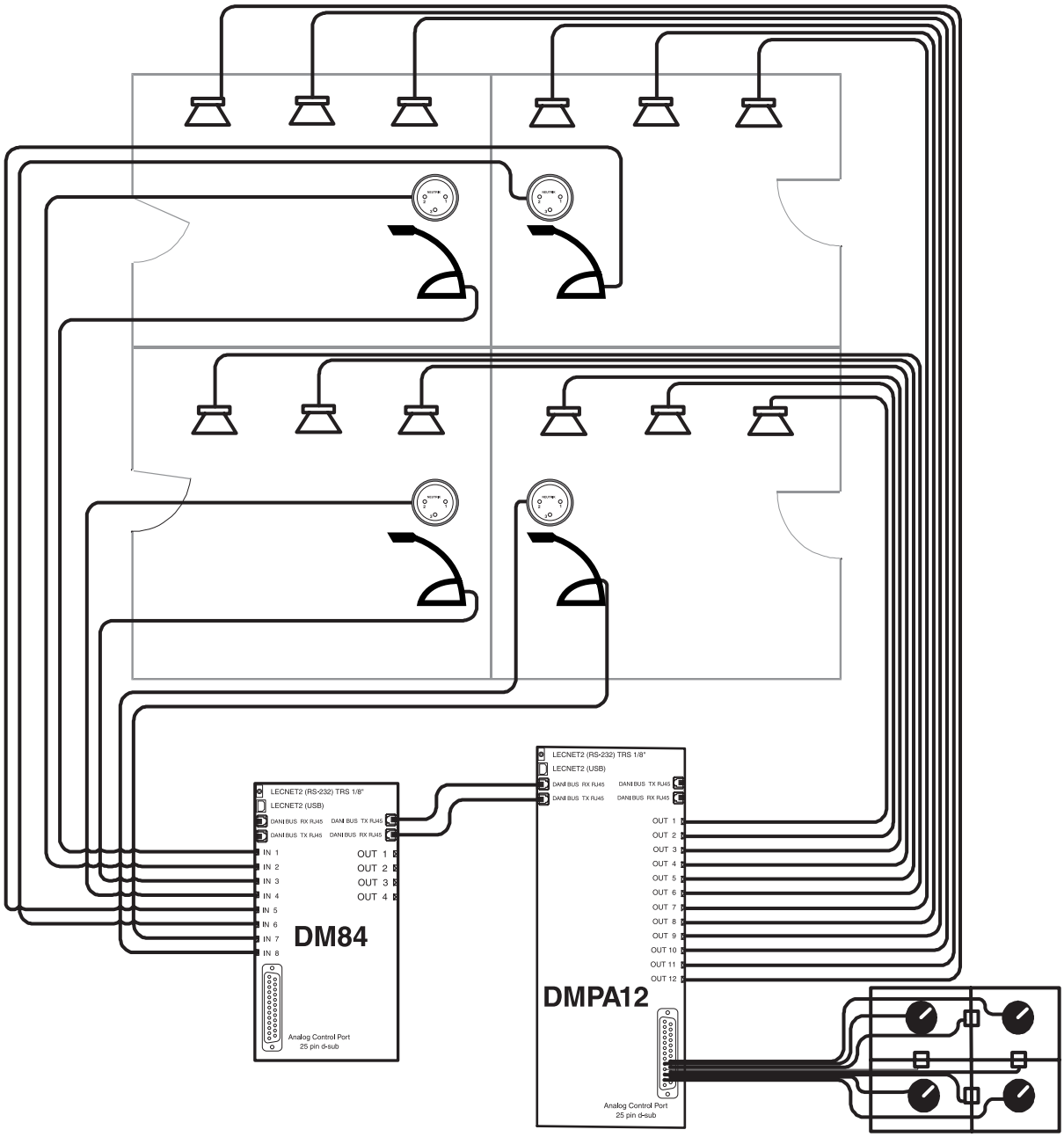
City Council Chamber



This is a representation of a typical City Council Chamber, including microphones for 16 council members, podium, tables and wireless systems. Ample inputs are provided for various audio-visual sources. Multiple PA zones feed the main chambers along with the audience seating areas. Press feeds are provided as is an output for an assistive listening system.

Any number of additional inputs can easily be added by expanding the system with one or more DM matrix mixers. Up to 24 presets and 128 macros can be used for multiple setups and user scenarios.

Multi-Room Combining



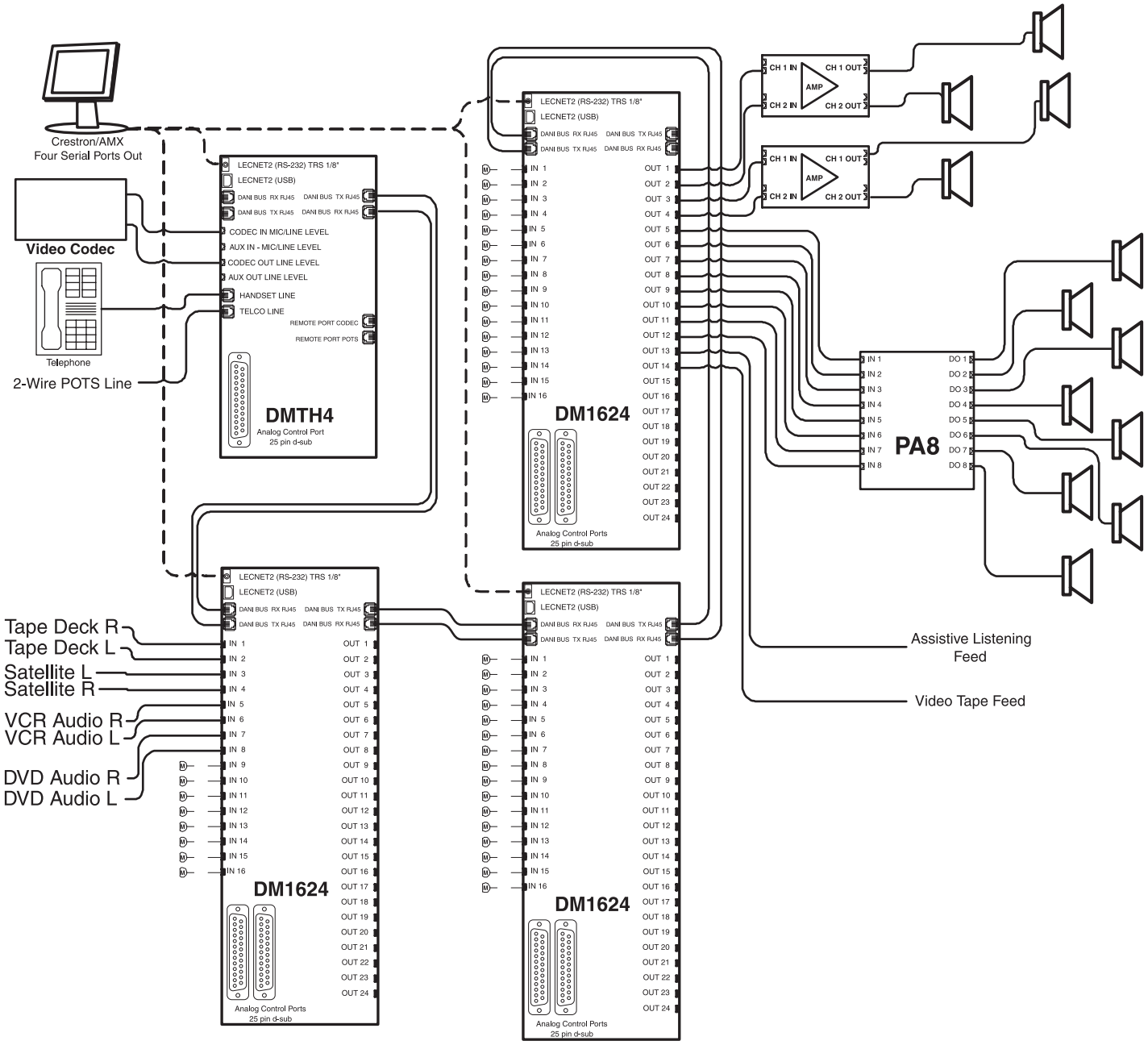
Analog Remote Control Panel

This four-room combining system features one microphone and one auxiliary input per room. The DM84 provides the automatic mixing function, while the DMPA12 amplifier feeds each of the 12 loudspeakers. Control of the system is via an analog control panel either in one of the rooms or located elsewhere. Automatic mixing and NOM attenuation is preserved in each room. Background music can be individually selected and paging signals can receive priority over local sources.

Larger systems can be configured with additional DM units connected via the expansion ports. For example, two DM84 mixers can provide 16 inputs with eight outputs. An alternative would be to choose a larger DM matrix mixer, such as a DM1612 or DM1624.

See companion disk for macro examples.

Collegiate Distance Learning/Multi-Purpose Hall



This example illustrates a large distance learning room with forty microphones, three-site conferencing (local, Video, and Telco), and several multi-media inputs.

The design provides extensive signal processing, including EQ, compression, delay, automatic feedback suppression, automatic microphone mixing, mix-minus speaker zoning and stereo playback over the main Program speakers. A large gallery audio feed with time delay compensates for the difference in distance from the speakers near the freshmen seats in the back of the room to the front program speakers.

All of this is possible with just three DM1624 mixers, power amplifiers, and speakers. No additional signal processing devices are required.

The touch screen control system can be easily programmed using simplified serial command strings and up to 128 resident macros. Up to 24 presets allow global recall of settings for alternate setups.

Calculating PAG, NAG and POWER

The PAG-NAG Computer Program

The GAINCALC software is a proprietary program offered by Lectrosonics on our CD ROM and by download from the web site as a part of the LecNet2™ sound system Designer's Kit. The supplied software includes EPS graphic files of LecNet2™ component control panels, block diagrams in DXF format of the sound system examples illustrated in this guide, and engineer's and architect's specifications in TXT format for all LecNet2™ components. The GAINCALC program (which runs under Windows® 98/NT/XP*) is designed to automate calculations of PAG, NAG and loudspeaker electrical power requirements for any type of indoor sound system.

The following are descriptions of each of the three parameters calculated by the program, and some information about how those calculations are made.

What is NAG?

Needed Acoustic Gain (NAG) is a measure of how much reinforcement a sound system must provide so that distant listener can hear a talker at a sound level comparable to when the listener is near the talker. As an illustration, assume that a listener near the talker experiences an average sound pressure level (SPL) of 75dB in normal conversation without sound reinforcement.

Next, assume that a more distant listener hears the same conversation at an average level of 62dB. This level would be low enough that intelligibility could be a problem, particularly in the presence of sound background noise. For the distant listener to hear normal conversation at the same average level as the nearby listener (i.e. 75dB SPL), an extra 13dB is needed at the distant listener's position. This is the Needed Acoustic Gain (NAG) for this example.

In order to make a NAG calculation using GAINCALC, the boxes labeled Dm, D0, Ld, and Lr must be filled in. The formula for calculating NAG is:

NAG = Lr-Ld-20Log10(Dm/D0)

As should be clear, NAG is a function of the physical distances between talkers and listeners. As yet, nothing has been said about a sound system. If the NAG value is positive, which it generally is, a sound system will be needed to provide acoustic gain at least equal to the NAG for the distant listener in order that they hear the same SPL as the nearby listener. This leads to the PAG calculation.

Download the program FREE from the web:

www.lectrosonics.com

What is PAG?

Potential Acoustic Gain (PAG) is a measure of how much extra reinforcement (acoustic gain) the sound system can be expected to provide for a distant listener above the level at which that listener would hear the talker without any sound reinforcement. Following through from the NAG calculation above, the PAG of a system should be at least equal to the NAG in order to provide sufficient SPL to the distant listener.

The PAG calculation is based on the assertion that the SPL generated by the sound system at the talker's microphone can cannot exceed the SPL that the talker produces acoustically at the same microphone. If the reinforced SPL exceeds the original SPL from the talker, the system regenerates, producing what is commonly known as FEEDBACK. The closer the loudspeaker is to the microphone, or the farther away the talker is from the microphone, the lower the PAG of the system will be. The other factor in the PAG calculation is the distance of the distant listener from the nearest loudspeaker. The further the listener is from the speaker, the lower the system PAG.

In order to make a PAG calculation using GAINCALC, the boxed labeled D0, D1, D2 and Ds must be filled in. The formula for calculating PAG is:

PAG=20Log10((D1D0)/(D2Ds))

Note that the PAG calculated from this formula implies a sound system which is right on the verge of feedback. In the Options menu of GAINCLAC, you can check FSM (Feedback Stability Margin) compensation. FSM compensation will subtract 6dB from the calculated PAG value to give a more realistic indication of PAG. More explanation of what can be done to maximize PAG can be found in the GAINCALC Help file.

What About Loudspeaker Power?

After the NAG and PAG calculations have been made, it's helpful to known how much amplifier power will be needed to produce the desired sound system SPL. GAINCALC uses the sensitivity data of the system loudspeaker(s), the distance between the distant listener and the closest loudspeaker, and the desired SPL at the distant listener's position to calculate the needed amplifier power. Note that GAINCALC adds 20dB to the desired SPL factor Lr in order to account for peak speech levels without clipping. One result of this is that if you require high SPL at the distant listener's position, you'll find yourself needing enormous amounts of amplifier power.

In order to make a power calculation, you should fill in the boxes labeled Spr Sens (db), @ Power (W), @ Distance, # of Speakers, Lr, and D2. The formula for the power calculation is:

Power = # of speakers x10a

Where a= (Lr+20 SpkrSens 20xLog(SpkrDist/D2))/10

PAG-NAG Software GUI

The program operates in two different scenarios, one for multiple loudspeaker systems such as in a board-room (upper illustration) and the other for cluster or central loudspeaker systems in larger spaces (lower illustration). Values are entered into the data entry cells for distances between microphone, talker, loudspeakers and listeners, and the targeted SPL at the listener's ears is selected.

A performance meter is also provided that indicates an approximation of how well the system would work with the given values. The meter updates continuously as the different values are entered. The program is a very valuable aid early in the design phase of a sound system project in determining the placement and quantity of loudspeakers and microphones.

Introduction to Writing Macros for LecNet2

One of the most powerful features of the new **DM Series** matrix mixers is the macro scripting language. The DM Series products contain 128 global macros; furthermore, each macro can hold up to 64 serial commands. If needed, macros can be chained if a long sequence of commands is required.

There are three types of instructions that can be contained in a macro: **Query**, **Update** or **Command**. A *Query* will always have a question mark (?) and an *Update* will always have an equal sign (=). The *Command* will have neither a question mark nor an equal sign.

In General

The syntax used for LecNet2 instructions closely follows natural English in order to make it easy to use. For instance, to set the input gain for channel 5 on a DM unit to 0dB, we might say "Input gain of channel 5 equals 0dB". This is a simple, understandable expression of the desired update. However, typing the above expression might become cumbersome if a large number of instructions were required in a LecNet2 application. Thus, the macro language is simplified to be quick to type and yet still easy to understand.

To create the actual instruction line, first we abbreviate "input gain" to "ingn" and shorten "channel 5" to simply "(5)". Thus we now have ingn(5). By adding the new gain value to the input channel, the complete Update instruction is now:

```
ingn(5)=0
```

Each instruction must then be followed with a carriage return, notated as <CR>.

Query

In order to get specific information from a DM mixer, the **Query** instruction is used to request the specific value needed. For instance, to determine the output gain value of channel 22, the instruction would thus be:

```
outgn(22)?<CR>
(the CR represents Carriage Return)
```

This would return an OK followed by the value. If the syntax of the query is not correct, the ERROR message will be returned. It is also possible to query all the channels with a single instruction by using the asterisk (*) character as a wild card:

```
outgn(*)? <CR>
```

This would return: OK {0,0,0,0,0,0,0,0,0,0,0,0} if all the output gain values were set to 0dB and the DM mixer was either a DM812 or a DM1612 (i.e. each has 12 outputs). For the DM1624 (24 outputs), 24 values would be returned.

Update

In order to change the input gain of a particular channel, an **Update** would be sent, including the channel number and the value. For instance, to set the output gain of channel 7 to -6dB, the instruction would be:

```
outgn(7)=-6 <CR>
```

In the case where it is required to update 14 of the 16 inputs (such as for a DM1612 or DM1624), there is no need to re-type the Update instruction 14 times. Instead, simply use this instruction:

```
ingn(*)={0,0,0,0,0,0,0,0,0,0,0,0,0,0, 99,99}
```

The asterisk indicates that all input channels will be addressed; the 16 values in the brackets represent the 16 gain settings to be sent to the DM mixer. The first element is input channel 1 and the last element is input channel 16. Channels 1 through 14 will be set to 0dB; channels 15 and 16 will be left untouched as indicated by the 99s in positions 15 and 16. "99" is the "don't care" value for the "ingn" instruction. It is often the case that a mass update should leave one or more values untouched, so "don't care" values are specified for many commands. See the online help for a particular command to learn more.

Command

If a macro or a preset needs to be invoked, a **Command** instruction is used. Unlike an Update, a Command will never set a value. And unlike a Query, a Command will never ask for a value. For example, to call preset 1, the instruction is simply:

```
recall(1)<CR>
```

Notice that neither a question mark nor an equal sign are present. Similarly, the instruction invoking macro 17 would be:

```
run(17)<CR>
```

LecNet Command Terminal

The LecNet2 command language has many types of instructions that are fully documented with examples in the **User Manual** and also in the **Help** section of the control panel of all LecNet2 devices. A good way to become familiar with these instruction sets is to run the **LecNet2 Command Terminal** found in the LecNet2 software interface. Once the **Command Terminal** has loaded and the DM device is connected via USB or RS232, type:

```
id?<CR>
```

The DM device will respond with OK {DMxxxx} (the Xs represent "1624, 1612, 812 or 84").

With this done it is possible to type in any of the documented instructions and view the real-time changes in the LCD on the front panel certain DM Series devices. For example, power on a DM1624 mixer and make sure it is connected to your PC. After the front panel shows DM1624 by Lectrosonics, push the "**Menu Select**" rotary control and then select **Setup**. Push **Menu Select** again and now select "**Inputs**". Next, push **Menu Select** once more to arrive at the **Input Setup** screen for channel 1. Push the soft key below "**Gai**". Now the input gain for channel 1 can be monitored in real-time.

Go back to the LecNet2 Command Terminal and type the instruction:

ingn(1)=-7<CR>

The LCD on the DM1624 front panel will immediately reflect the new value of –7dB for channel 1.

Creating Macros in LecNet2

There are several platforms usable to write and edit macros for LecNet2 device control. Macros may be written line-by-line in the **LecNet2 Command Terminal**, externally via a Windows®- or Mac-based text editor like **NotePad** or in the **Macro Editor** found in the LecNet2 Control Panel.

For the Macro Editor within the DM Control Panel, go to the **Macros** menu and select **Macro Editor**. Give the Macro a name (such as "Crosspoint 1-3") and click in the single-line field **Commands to Execute**. As an example, enable three crosspoints with the following set of instructions:

```
xpgn(1,1)=0<CR>
xpgn(2,2)=0<CR>
xpgn(3,3)=0<CR>
```

You have created a Macro titled **Crosspoint 1-3** containing three commands which enable three crosspoints in the matrix.

The next step is to save the macro to the DM mixer. While still in the Macro Editor, go to the "Device" menu and select "Store to device memory". Enter **1** as the number for this macro (normally you can choose a number between #1 through #128). If that macro location is empty, the **Title** field will be blank. If there is already a macro at that location, the title associated with that number will display. At that point, either choose a new number or over-write the old macro with the new one. Do not try to type a title in this field it is for reference only. Once you choose a macro number click **OK**. At this point the macro is written to the DM mixer. Click **Done** to close the **Macro Editor**.

In order to test the newly written Macro, go to the **Macro Editor** menu and choose **Run Macro**. Select the macro you would like to run by title and number (choose **Crosspoint 1-3**). Click **OK**, then go to the Matrix tab and the three crosspoints will now be enabled in the Matrix.

As an additional example, create a second macro following the above instructions, using the same xpgn commands, except this time set them to –70 instead of 0. Give this macro a new name and save it to macro #2. Run the new macro and all of the crosspoints will be disabled in the matrix (-70dB is the equivalent of "off"). There are now two macros created and saved that will do the opposite of each other (toggle), located in macros #1 and #2 in the DM mixer.

Assigning Macros to External Buttons

To practice assigning our new macros to external SPST momentary switches, a **Test** button is provided in the DM mixer control panel that will simulate external switch contacts. In the control panel, choose the **Rear Panel Ctrl** tab. Select **programmable Input #1** (it might be selected already as a default). Assign the function to **Run Macro on Close**. Macro #1 will appear with the title that you assigned to it. Also, the programmable input tab (in the lower left) now shows **(RM)** – this is simply a handy reminder that it is assigned to run a macro.

Next, click the **Test** button, navigate to the **Matrix** tab and you will see the changes to the matrix due to the "contact" of a simulated external button. Now, select the tab for **Programmable Input #2** and assign it to run macro #2 in the same way you assigned programmable input #1. Test it and look at the matrix. You now have two simulated external buttons that will affect the matrix with the 3 commands that are assigned in the macros.

Advanced Macros

The material in the above section describes how two buttons can be used to toggle multiple crosspoints. In the next example, we will assume that the client requires the same functionality but only with a single button. To begin, we must modify Macros #1 and #2 with one line of code.

Open the **Macro Editor** in the **Device** menu and select **Recall from Device Memory**. Select Macro #1 and click **OK** to see the code appear in the edit window. Click in the **Commands to Execute** field and type:

```
prgindef(1) = {14,2} <CR>
```

Next, go to the **Device** menu and select **Store to Device Memory**. Save it to macro #1, thus replacing your old macro with the new modified version.

This command, prgindef(1), allows you to assign any of the 17 available library functions from the drop-down item list, such as **Run Macro on Close**, **Toggle Mute Inputs**, **Momentary Mute Inputs** and so on, dynamically. In the help file, note that #14 is **Run Macro on Close**, thus {14, 2} reassigns programmable input #1 to run Macro #2 the next time the button is pressed.

Next, modify Macro #2 with the code:

```
prgindef(1) = {14,1}<CR>
```

This will re-assign programmable input number 1 to run Macro #1 during the next time it is used. We now have a button that toggles between two Macros. Just as we modified Macro #1 above, go to macro #2 and add the line:

```
prgindef(1) = {14,1}<CR>
```

Save this change as before.

To confirm that the single button toggles between the two macros each time it is pressed, push the **Test** button for programmable input #1. Notice that it re-assigned itself to run Macro #2. Press the **Test** button again; it now has re-assigned itself to run Macro #1. After each button press, check the matrix and watch the toggle take place.

With this information, it should be clear how much power and control is available in the DM series of DSP matrix processors with very simple instructions and macros.

A Brief History of Product Development

Founded in 1971, Lectrosonics began with the manufacture of portable sound systems sold under the Voice Projector© registered trademark. The first product line consisted of a self-contained lectern/sound system and two over-the-shoulder portable sound systems.

In 1975 the first wireless microphone systems were introduced to audio visual markets. The first system was a lavalier system consisting of a belt-pack transmitter and matching receiver with narrowband IF filters called UNICHANNEL®. In keeping with a "total portability" concept, the first self-contained speaker/amplifier/wireless system was developed during the same period. The first VHF high band wireless microphone systems were introduced in 1987, taking the proven UNICHANNEL© design to a higher frequency band. The product line expanded rapidly over the following 5 or 6 years, leading to the introduction of UHF wireless systems in 1993. The first frequency adjustable synthesized UHF wireless system began shipments in 1995, followed by a compact version of the receiver and a complement of belt-pack, plug-on, and hand-held transmitters.

In March 1998 the first wireless IFB (Interruptible FoldBack) systems were shipped. This was the first UHF system on the market. Being a synthesized UHF design with extended operating range and excellent audio performance, the IFB Series was an immediate success.

Advanced DSP-based wireless technology was introduced to the market with Digital Hybrid Wireless™ systems in late 2002. The technology combines 24-bit digital audio with analog RF to eliminate compandor artifacts and preserve the operating range and spectral efficiency of the finest analog wireless systems. The DSP also provides compatibility with analog systems from Lectrosonics and other manufacturers. A patent application was submitted prior to shipments in 2002 and has been pending through the date of this document.

Development in wireless microphone systems continued with the introduction of an encrypted digital system in early 2003, following several years of R&D. Introduced as the 700 Series, with a 19" half rack diversity receiver, belt pack and hand held transmitters, the system is presently in use by a variety of security conscious private companies in addition to several federal government departments.

In 1989 the first automatic mixer products were introduced in a modular configuration called the Modular Audio Processor. This was a "card cage" design that held up to ten input or signal processing cards allowing the customization needed for medium to large scale sound systems in courtrooms, boardrooms, council chambers, conferencing, lecture halls and worship centers. Numerous signal processing modules were added to this product group over the next five or six years. As the popularity of automatic sound systems continued to grow and competition grew more fierce, the appeal of microprocessor control turned into necessity, leading to the first LecNet™ components in late 1994.

LecNet™ was launched In December 1994, as the second generation of audio products to address an increasing demand for fully automatic sound systems in courtrooms, lecture halls, council chambers, teleconferencing and distance learning applications. LecNet components communicated with each other via on-board microprocessors, with a host PC used for setup and monitoring. The automatic mixers in this group utilized a patented Adaptive Proportional Gain Algorithm* to apply NOM attenuation without switching and to prevent background noise from affecting the mixing action. The LecNet family grew from the first stand-alone automatic microphone mixer to larger automatic matrix mixers, multi-channel DSP processors, a digital telephone hybrid, an 8-channel power amplifier, and a variety of accessories and remote controls.

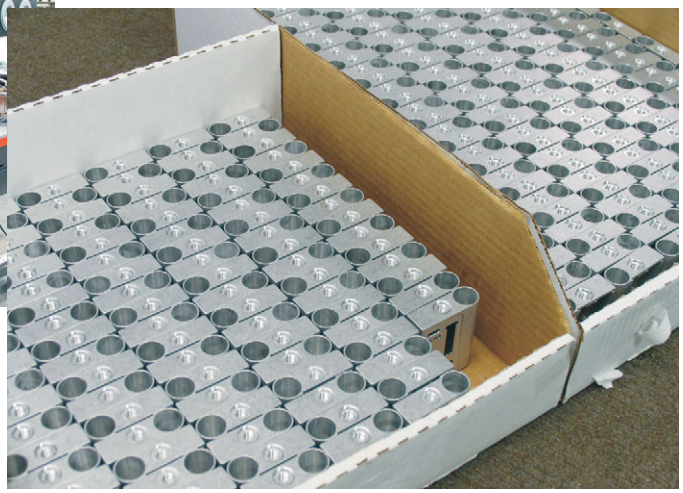
In November 2004, the first shipments of LecNet2 DM Series digital matrix processors took place. Three models were offered: 16in/24 out, 16in/12out and 8in/12out. In addition to a digital crosspoint matrix, a DANI (digital audio network interface) bus was introduced. This fully digital bus allowed audio and control signals to flow upward from slave to master units, then return from the master unit backward to the slaves. This new concept called "back propogation" expanded the flexibility and signal routing cabability beyond the earlier analog designs. Extensive signal processing was also available on every input and output.

The DMTH4 digital telephone hybrid was added in early 2006, followed by the DMPA12, a 12-channel digital power amplifier that tied directly into the DANI bus via CAT-5 connections on the rear panel. The DM84 model was introduced in June 2006 to replace the earlier analog models with 8 inputs.

Extensive control flexibility of LecNet2 components was built into the basic architecture. AMX® and Crestron® control systems* easily integrate with LecNet2 components, with a powerful macro language added to further simplify programming and control options.

Lectrosonics remains an engineering driven company. Ongoing efforts continuously produce new designs with the latest technology in electronics and mechanical engineering.

** AMX® is a registered trademark of AMX Corporation.*
Crestron® is registered trademark of Crestron Electronics, Inc.



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